

TITLE OF THE INVENTION

METHOD AND APPARATUS FOR CONCURRENTLY
ESTIMATING RESPECTIVE DIRECTIONS OF A PLURALITY OF
SOUND SOURCES AND FOR MONITORING INDIVIDUAL SOUND
5 LEVELS OF RESPECTIVE MOVING SOUND SOURCES

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates to a method and
apparatus for estimating the respective angular
10 directions of one or more sound sources in relation
to a specific position, and to a method and
apparatus for using such estimated directions to
detect when a moving sound source has moved through
a specific angular direction and to monitor the
15 sounds emitted from respective ones of a plurality
of moving sound sources.

2. Description of the Prior Art

One method of estimating the angular direction
of a sound source and of monitoring a sound source
20 is described in Japanese patent HEI 5-114098. Fig.
22 shows the configuration of this prior art sound
source direction estimation apparatus. In Fig. 22,
first and second directional microphones 901 and 902
are set up spaced apart by a fixed distance L, along
25 a line of flow of traffic. The sounds which are

09744716 2/4/80

respectively gathered by these microphones, which
can be referred to as traffic noise A and traffic
noise B, are respectively amplified by amplifier
circuits 903, 904, and the resultant signals are
5 alternately selected by a changeover circuit 905,
under the control of a timing control circuit 910,
to be successively subjected to frequency analysis
by a frequency analysis circuit 906. Frequency
vector distributions SA and SB are thereby
10 respectively obtained, corresponding to traffic
noise A and traffic noise B. Next, the degree of
similarity between the frequency vector
distributions SA and SB is detected, and the time
difference dt between the points at which the
15 frequency vector distributions SA and SB become
approximately identical is obtained, by a time
difference detection circuit 908. A time
difference/velocity converter circuit 909 then
performs the calculation:

$$V = L/dt$$

20 to determine the velocity of the sound source, and
displays this velocity value by a display circuit
911. In addition, the direction of the sound source
can be calculated, based on the aforementioned time
25 difference. In that way, with this prior art

method, it is possible to estimate the angular direction of a sound source and to monitor the sound emitted from that sound source.

However with such a prior art type of sound source direction estimation method, when sounds from a plurality of sources are simultaneously reaching the microphones, or when there is a sound source other than the sound sources which it is desired to monitor, then the accuracy of estimating the direction of a sound source will become low.

In the following specification and appended claims, the term "direction" is to understood as signifying "angular direction with respect to a specific origin point", unless used in referring to a direction of angular motion or direction of linear motion.

SUMMARY OF THE INVENTION

It is an objective of the present invention to overcome the above problems of the prior art, by providing a method and apparatus for sound source direction estimation which enables the direction of a sound source in relation to a specific point to be accurately estimated even when there is a plurality of sound sources in the vicinity, or a sound source

of a type other than other than desired sound sources.

It is a further objective to provide such a method and apparatus whereby respective directions of a plurality of sound sources can be concurrently estimated.

It is moreover an objective to provide a method and apparatus for utilizing these estimated directions to detect when a sound source has moved through a specific direction in relation to a fixed position, and thereby detect that the sound source has moved past that specific position.

It is moreover an objective to provide a method and apparatus for utilizing these estimated
15 directions to detect whether a sound source is in motion, and if so, the motion direction and velocity.

It is moreover an objective to provide a method and apparatus for utilizing these estimated
20 directions to enable the sound emitted from a specific sound source to be monitored.

To achieve the above objective of obtaining estimated directions of one or more sound sources, with a sound source direction estimation method and apparatus according to the present invention, the

input signals of an array of microphones are
extracted in each of respective time windows, while
in each of the time windows, respective sound source
estimated directions are calculated for each of a
5 frequency of a fixed set of frequencies, and the
average of the directions thus calculated becomes
the estimated direction corresponding to that time
window. Since a sound source estimated direction
is obtained based on a plurality of frequencies, a
10 high degree of sound source direction estimation
accuracy can be achieved.

To further increase the accuracy of sound
source direction estimation, as each frequency-
averaged estimated direction is obtained as
15 described above for a time window, the average of
the estimated direction obtained for the time window
and the respectively estimated directions which have
been calculated for a fixed plurality of time
windows which precede that time window is
20 calculated, to thereby obtain (for each of
successive time windows) an estimated direction
which is based on both frequency-averaging and time-
averaging.

Furthermore, with such a sound source direction
25 estimation method and apparatus according to the

present invention, a judgement can be made as to whether the direction of a sound source has passed through a specific direction, and data expressing a passage detection result then generated to indicate that fact, by detecting whether a number of estimated directions obtained for that sound source which are each within a fixed range of directions and occur within a fixed observation interval exceeds a predetermined minimum number. In that way, it becomes possible to detect that a moving sound source is currently passing the microphone array.

Furthermore, with such a sound source direction estimation method and apparatus according to the present invention it becomes possible to judge the direction of motion of a sound source, when a passage detection result corresponding to the sound source is generated, based upon the sign of the difference between estimated directions respectively obtained for the sound source before and at the time point at which the passage detection result is generated. For example, successive sets of obtained estimated directions can be temporarily set into a data buffer, so that when a passage detection result is obtained, the buffer contents will contain estimated directions which were obtained prior to

the point of generating the passage detection result. Alternatively, it may be possible to achieve a similar effect by using an estimated direction which is obtained subsequent to the time at which the passage detection result is obtained.

Furthermore, if the distance between the microphone array and a line of motion of a stream of moving sound sources is known, then the estimated directions can be used to estimate the speed of (linear) motion of a sound source, using the time point at which a passage detection result corresponding to that sound source is generate. Specifically, the time taken for the estimated directions obtained for that sound source to move through a specific amount, or the amount by which the estimated directions change within a predetermined time interval, can be measured to thereby obtain the angular velocity of the sound source in relation to the microphone array. The approximate linear velocity of the sound source can thereby be calculated based on the distance to the line of motion of the sound source.

Furthermore, with a method and apparatus according to the present invention, it becomes possible to arrange that when it is detected that a

sound source is currently moving past the microphone array (as indicated by a passage detection result), a directivity is established for the microphone array, aligned along the most recently obtained estimated direction of that sound source. If the microphone array is a fixed linear array, then a directivity within a range of approximately 180° can be selectively established by applying appropriate amounts of delay to the respective microphone output signals, and summing the delayed signals, with the resultant signal constituting a monitoring signal which conveys the sound being emitted from that specific sound source.

Alternatively, a plurality of fixed directivities for the microphone array can be predetermined. In that case, when a passage detection result is obtained for a sound source, the current estimated direction of the sound source can be used to select the one of the predetermined directivities that is closest to that estimated direction, and monitoring of the sound emitted from that sound source then performed. In that case it becomes possible to monitor sound sources which are moving at various substantially different velocities (and hence may vary substantially with respect to

the estimated direction which is obtained for each sound source at the time point when passage of the sound source is detected) since it becomes unnecessary to establish a new microphone array
5 directivity each time that monitoring is to be started.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a conceptual diagram for use in describing the basic features of a first embodiment,
10 for successively deriving respective estimated directions of one or more sound sources in relation to a microphone array;

Figs. 2A 2B constitute a general system block diagram of a first configuration of a sound source
15 direction estimation section of the first embodiment, whereby estimated directions are derived in successive time windows by using both frequency-based and time-based averaging processing;

Figs. 3A, 3B constitute a flow diagram of a
20 processing sequence executed by the first configuration of the sound source direction estimation section;

Fig. 4 constitutes a general system block diagram of a second configuration of the sound
25 source direction estimation section of the first

embodiment, whereby estimated directions are derived in successive time windows by using only frequency-based averaging processing;

5 Figs. 5A, 5B constitute a flow diagram of a processing sequence executed by the second configuration of the sound source direction estimation section;

10 Fig. 6 is a general system block diagram of a second embodiment, having a sound source direction estimation section and a passage detection section for detecting when a moving sound source has passed through a specific direction in relation to a microphone array;

15 Fig. 7 is a flow diagram of a processing sequence executed by the second embodiment;

20 Fig. 8 is a general system block diagram of a third embodiment which includes a sound source direction estimation section, a passage detection section, and which derives and provides to a recording apparatus data expressing a level of sound power received from a specific sound source which is currently moving past a microphone array;

25 Fig. 9 is a flow diagram of a processing sequence executed by the third embodiment;

Fig. 10 is a general system block diagram of a fourth embodiment which includes a sound source direction estimation section, and serves to detect the presence of a stationary sound source;

5 Fig. 11 is a general system block diagram of a fifth embodiment, which includes a sound source direction estimation section, a passage detection section, and a motion direction derivation section which derives data expressing the direction of motion of a sound source which is currently moving past a microphone array;

Fig. 12 is a flow diagram of a processing sequence executed by the fifth embodiment;

10 Fig. 13 is a general system block diagram of a sixth embodiment, which includes a velocity detection section for operating on passage detection results obtained by a passage detection section and estimated directions obtained by a sound source direction estimation section to obtain the velocity of a sound source which is currently moving past a microphone array;

Fig. 14 is a flow diagram of a processing sequence executed by the sixth embodiment;

15 Fig. 15 is a general system block diagram of a seventh embodiment, which includes a directivity

25

control section for operating on passage detection results obtained by a passage detection section and estimated directions obtained by a sound source direction estimation section, for setting the
5 directivity of a microphone array such as to monitor a passing sound source;

Fig. 16 is a flow diagram of a processing sequence executed by the seventh embodiment;

10 Fig. 17 is a general system block diagram of an eighth embodiment, which includes a directivity control section for operating on passage detection result and estimated directions to select one of a plurality of predetermined directivities of a microphone array for monitoring the sound emitted
15 from a passing sound source;

Fig. 18 is a flow diagram of a processing sequence executed by the eighth embodiment;

Fig. 19 is a conceptual diagram for use in describing the operation of the embodiment of the
20 eighth embodiment;

Fig. 20 is a conceptual diagram for use in describing a modified configuration of the eighth embodiment, for monitoring sound levels of sound sources moving along two adjacent paths with
25 opposite directions of motion;

Fig. 21 is a partial flow diagram corresponding to Fig. 20; and,

Fig. 22 is a general system block diagram of a prior art type of apparatus utilizing microphones to
5 detect the directions of moving sound sources.

DESCRIPTION OF PREFERRED EMBODIMENTS

First Embodiment

The basic features of a first embodiment of the invention will first be described referring to the conceptual diagram of Fig. 1. In Fig. 1, numeral
10 101 denotes a sound source such as a motor vehicle (as seen in plan view) travelling along a path 95 in the direction indicated by the arrow, for example along a road 90. A linear array of a plurality of
15 microphones 102 is disposed adjacent to the path 95 with the array direction parallel to the path. Respective output signals from the microphones of array 102 are supplied to a sound source direction
20 estimation section 116, in which the signals are processed to detect successive directions (expressed with respect to a specific point as origin, such as the location of a central microphone of the microphone array 102) of each of one or more sound
sources such as the sound source 101 which are

moving in the arrow direction along the path 95 within the indicated direction estimation range.

Each direction of a sound source will be assumed to be expressed as an angular displacement from a central direction (the direction indicated as "0°" in Fig. 1), with each direction expressed as a positive or negative angular displacement with respect to that central direction. As noted above, this use of the term "direction" when used alone, referring to an angular value, is to be distinguished from the term "direction of motion" of a sound source, such as the linear motion direction indicated by the arrow for the sound source 101 in Fig. 1.

The sound source direction estimation section 116 is capable of obtaining estimated directions of a sound source within a range which might for example be from +90° to -90°, or from +45° to -45°. However that range of directionality (in azimuth) may be restricted in order to eliminate extraneous noise sources, and in addition the range of directionality in elevation may also be restricted, for the same reason.

The configuration of the first embodiment will be described in greater detail referring to the

general block diagram of Figs. 2A, 2B. Here, a sound source 101 and microphone array 102 correspond to the identically numbered items shown in Fig. 1. In Figs. 2A, 2B, the microphone array 102 is formed of M microphones which are spaced apart at regular spacings d. The respective output signals from the array of microphones 102 are transferred through M waveform extraction sections 103 of a sound source direction estimation section 116 to be supplied to M frequency analyzers 104 respectively. For each frequency of a predetermined set of frequencies, a corresponding complex amplitude matrix is derived by the frequency analyzers 104. A correlation matrix calculation section 107 derives, for each such complex amplitude matrix, a corresponding correlation matrix. Each such correlation matrix (i.e., each matrix corresponding to a specific frequency value) is operated on by an eigenvector calculation section 108 to calculate a corresponding set of eigenvectors. The eigenvector calculation section 108 supplies the calculated eigenvectors to a noise component matrix calculation section 109, and each noise component matrix which is thereby derived (again, corresponding to a specific frequency) is supplied to a direction/power

relationship calculation section 110. A plurality of direction control vectors are also supplied to the direction/power relationship calculation section 110, from a direction control vector calculation
5 section 106.

Output values produced from the direction/power relationship calculation section 110 are supplied to an out-of-range value exclusion section 112, together with data expressing a predetermined range
10 of directions, produced by a direction range setting section 111. Output values produced from the out-of-range value exclusion section 112 are supplied to a frequency basis averaging section 113, and resultant output values are supplied to a time basis
15 averaging section 114. Resultant output values produced from the time basis averaging section 114 express respective estimated directions of sound sources, i.e., these constitute the output data which are generated by the sound source direction
20 estimation section 116.

The operation of this embodiment is as follows. The microphone array 102 is formed of M microphones which are arranged at regular spacings d along a straight line. The output signals from these M
25 microphones are periodically extracted in respective

time windows, each time window having a duration W ,
by the waveform extraction sections 103. The time
window shape can be rectangular, or alternatively a
shape such as that of a Hamming window could be
5 used, in which the amplitude is made smaller at the
opposing ends of the window. If the time window
length W is made relatively small, then the accuracy
is lowered, while if W is made relatively large then
it becomes less possible to follow a sound source
10 which moves rapidly. Thus the time window length W
must be appropriately selected in accordance with
the anticipated speed of motion of the sound sources
whose sounds are to be monitored. For example, in
the case of estimating the direction of a sound
15 source which moves at a velocity of 40 km per hour
and whose motion path is spaced apart from the
microphone array 102 by a distance L which is equal
to 10 meters, then a suitable value for the time
window length W is in the range 2 to 10 ms. In
20 addition, a suitable value for the extraction period
(i.e., repetition period of the time windows) would
be in the range $W/2$ to $2W$.

At each time window, the frequency analyzers
104 operates on each set of microphone signal
25 portions (which can be considered as respective

time-axis signal portions) that are extracted by the waveform extraction sections 103, to calculate a plurality of sets of complex amplitude values, with each of these sets corresponding to one of the
5 aforementioned predetermined set of frequency values. The FFT (Fast Fourier Transform) method can be used to calculate the complex amplitude values, or alternatively, if the number of frequencies for which the calculation must be performed is
10 relatively small, the DFT (Discrete Fourier Transform) method could be applied.

Next, for each of the aforementioned set of frequencies, the values of received sound power (as represented by received signal power) for each of
15 respective directions are calculated. If the wavelength of a frequency is less than twice the value of the spacing d between adjacent microphones of the microphone array 102, then the higher the frequency, the greater will be the estimation
20 accuracy. For that reason, in practice, frequencies having a wavelength value in the range $2d$ to $10d$ are suitable. For any specific frequency, the complex amplitude matrix is calculated, and this is expressed as a vector $X[m]$, by the following
25 equation (1).

$$X[m] = [x_1, x_2, \dots, x_M]^T \dots\dots\dots (1)$$

Here, x_m ($m = 1$ to M) are the respective complex amplitudes that are calculated for that specific frequency, based on the m output signals produced from the microphone array, and the superscript T signifies that $X[m]$ is the transpose of the matrix $[x_1, x_2, \dots, x_M]$. Next, equation (2) below is applied, to calculate the corresponding correlation matrix 107, expressed as the matrix $R[m, m]$:

$$R[m, m] = X[m] \cdot X[m]^H \dots\dots\dots (2)$$

where $m = 1$ to M . Here, the superscript H signifies the transposed complex conjugate. The eigenvector calculation section 108 then calculates the set of eigenvectors $v_1[m], v_2[m], \dots, v_M[m]$ (where m takes the values 1 to M) corresponding to the correlation matrix $R[m, m]$.

Since the matrix $R[m, m]$ is a Hermitian matrix, the calculation of the edge vectors can be performed by using the Householder method, which is known in this field of technology, to convert the matrix R to a symmetric tri-diagonal matrix, then applying the QL method (which is also well known in this field of technology) to obtain the eigenvectors.

The noise component matrix calculation section 109 then calculates the noise component matrix $R_n[m, m]$ corresponding to the noise components of the microphone output signals, by using equation (3) below, assuming that there are K sound sources, where K is an integer.

$$R_n[m, m] = v_{K+1}[m] \cdot v_{K+1}[m]^H + v_{K+2}[m] \cdot v_{K+2}[m]^H + \dots + v_M[m] \cdot v_M[m]^H \quad \dots\dots\dots(3)$$

The above equation assumes that the number of sound sources K is less than the total number of microphones in the microphone array 102, i.e., is no greater than $M-1$. If the number of sound sources cannot be estimated beforehand, then K should be set equal to $M-1$.

Next, to estimate the respective values of sound power received from respective directions, respective direction control vectors for each of a predetermined plurality of directions θ are calculated, where each of these is an angular direction, expressed in relation to a point on the microphone array. Each direction control vector is obtained as a column vector $d[m]$, from equation (4) below:

$$d[m] = [1, e^{-j\omega\tau}, e^{-j\omega^2\tau}, \dots, e^{-j\omega(M-1)\tau}]^T$$

• • • • (4)

here, τ is defined by equation (5) below, in which c denotes the speed of sound:

$$\tau = (d \sin \theta) / c \quad \dots\dots\dots (5)$$

5

The direction/power relationship calculation section 110 calculates, for each of the directions θ , the corresponding power value $P(\theta)$, by using equation (6) below.

10

$$P(\theta) = 1 / (d[m]^H \cdot Rn[m, m] \cdot d[m]) \quad \dots\dots\dots (6)$$

Typically, the set of directions θ will cover an angular range of -90° to $+90^\circ$.

15

In that way, respective values of received sound power corresponding to different directions of a sound source are derived, for one specific frequency out of the aforementioned predetermined set of frequencies.

20

Next the direction θ_{\max} , i.e. the direction for which the highest value of received sound power $P(\theta)$ is obtained, is determined. The out-of-range value exclusion section 112 excludes θ_{\max} from the estimated direction values if it is outside a range of values that is set by the direction range setting section 111. The direction range setting section

25

111 sets that range based upon the range of possible locations of sound sources which are to be detected by the apparatus, if the latter range is known beforehand.

5 By the procedure described above, using a frequency which is measured within a specific time window, an estimated direction of a sound source can be calculated with respect to that specific frequency. That procedure is repeated for each of
10 the aforementioned set of frequencies, for that same time window. The set of respective estimated directions θ_{\max} which are obtained for the entire set of frequency values are then averaged (i.e., by the frequency basis averaging section 113) to
15 thereby derive an estimated direction of a sound source, as obtained using one specific time window.

 As these estimated directions are successively derived by the frequency-based averaging section 113, they are stored in a buffer within the time-
20 based averaging section 114, which is capable of holding a predetermined number of estimated direction values. Each time an estimated direction is newly derived by the frequency-based averaging section 113, i.e., for one specific time window, the
25 set of estimated directions which are currently held

in the buffer are read out, and the average of these estimated directions together with the most recently derived estimated direction is calculated.

In that way, each estimated sound source
5 direction which is produced from the time-based averaging section 114 is obtained based upon results obtained from a plurality of successive time windows.

Thus with this embodiment of the invention, the
10 signals from an array of microphones are periodically extracted using time windows, within each of the time windows an estimated direction is derived by using an average of respective estimated directions obtained using a plurality of
15 frequencies, and as each such estimated direction is derived, it is averaged together with the estimated directions similarly obtained in a preceding plurality of time windows, to thereby obtain an accurate estimated sound source direction by
20 performing averaging in both the frequency and time domains.

In the above it has been assumed that an estimated direction is to be derived for only one sound source at a time, i.e., that in each time
25 window period a single maximum value of sound power

is to be detected, and the direction corresponding to that maximum value derived as θ_{\max} . However in general there may be a plurality of sound sources (e.g., motor vehicles) within the range of directionality of the microphone array 102 at one time. In that case, respective local maximums of values of received signal power in relation to direction can be obtained for each of such sound sources, i.e. from the series of results that are obtained by successively applying the above equation (6) to each of the directions θ . In that way, it becomes possible to obtain a plurality of estimated directions concurrently for one time window, e.g., as $\theta_{\max 1}$, $\theta_{\max 2}$, etc., as respective directions corresponding to each of the local maximums of sound power.

An alternative configuration of this embodiment will be described in the following, referring first to the general system block diagram of Fig. 4. This is formed of a microphone array 102, a waveform extraction sections 103, a frequency analyzers 104, a correlation matrix calculation section 107, a direction control vector calculation section 106 and a frequency-based averaging section 113, whose respective functions and configurations are as

described for the form of the embodiment shown in
Figs. 2A, 2b, together with a direction/power
relationship calculation section 208. This form of
the first embodiment differs from that of Figs. 2A,
5 2B in the following respects. Firstly, the
direction/power relationship calculation section 208
operates directly on each correlation matrix that is
produced from the correlation matrix calculation
section 107 for a specific frequency (in conjunction
10 with a set of direction control vectors which are
generated from the direction control vector
calculation section 106) as described hereinabove,
to derive a direction θ_{\max} , i.e., the direction for
which the highest value of received sound power $P(\theta)$
15 is obtained at that particular frequency. Secondly,
only frequency-based averaging of estimated
directions is performed, to obtain (for each time
window) a finally outputted estimated direction of a
sound source.

20 As described hereinabove, for each of the
aforementioned predetermined set of frequency
values, a corresponding complex amplitude matrix is
derived by the processing that is applied by the
frequency analyzers 104 to a set of extracted audio
25 signal portions from the respective microphones of

array 102, and this matrix can be expressed as a vector $X[m]$, as shown by the aforementioned equation (1). For each such complex amplitude matrix, the corresponding correlation matrix is then calculated
 5 by the correlation matrix calculation section 107, using equation (2) above, i.e., with the correlation matrix being expressed as $R[m, m]$. In addition, a set $d[m]$ of direction control vectors are derived by the direction control vector calculation section 106
 10 for a plurality of directions θ , by using equation (4) above, and supplied to the direction/power relationship calculation section 208.

The direction/power relationship calculation section 208 then uses the following equation (7) to
 15 calculate the sound power $P(\theta)$ at each of the plurality of directions θ for one specific frequency (i.e., directions covering an angular range which might be for example from $+90^\circ$ to -90°):

$$20 \quad P(\theta) = d[m]^H \cdot Rn[m, m] \cdot d[m] \quad \dots\dots\dots(7)$$

The direction θ_{\max} for which the maximum sound power level is obtained, at one specific frequency, is thereby determined.

The above procedure is repeated for each of the
 25 aforementioned predetermined set of frequencies.

The average of the respective directions θ_{\max} obtained for the various frequencies is then derived by the frequency basis averaging section 113, to thereby obtain (for a specific time window) the
5 estimated direction of a sound source.

The processing sequence corresponding to the above is shown in the flow diagram of Figs. 5A, 5B.

Second Embodiment

A second embodiment of the invention will be
10 described referring first to the general system block diagram of Fig. 6. In Fig. 6, a microphone array 102 and a sound source direction estimation section 116 have the respective functions and configurations described hereinabove for the
15 correspondingly numbered components of the first embodiment. The sound source direction estimation section 116 can have the first configuration shown in Figs. 2A, 2B, whereby each output estimated direction is derived based upon frequency-based
20 averaging and upon time-based averaging applied over a plurality of successive time windows, or can have the alternative configuration shown in Fig. 4, with only frequency-based averaging being applied. Data expressing respective estimated directions of a
25 sound source are supplied from the sound source

direction estimation section 116 to an in-range
occurrence number calculation section 212 of a
passage detection section 216, which also consists
of a direction range setting section 211 and a
5 passage detection judgement section 213. Data
expressing a predetermined range of directions which
will be designated as the passage detection range
Pn, produced from the in-range occurrence number
calculation section 212, are also supplied to the
10 passage detection judgement section 213. That
predetermined range of directions Pn is within, but
no greater than, the entire range (such as from
+90° to -90°) for which estimated directions can be
derived by the sound source direction estimation
15 section 116, and in Fig. 6 is shown as extending
from an initial direction θ_r to a final direction
 θ_f . Data expressing a passage detection result is
thereby obtained as output from the passage
detection judgement section 213, i.e., data
20 indicating that a sound source is currently passing
by the microphone array 102.

Specifically, the in-range occurrence number
calculation section 212 calculates, for a
predetermined plurality N of successive time windows
25 which begin when an estimated direction within the

range P_n is first obtained for the sound source, the number of times that estimated directions of the sound source are within the passage detection range P_n . Designating that number of times as n , if n exceeds a predetermined reference level n_t , then the passage detection judgement section 213 judges that a sound source has passed through the range P_n (more specifically, has passed through at least the initial direction of that range) and hence produces data expressing a passage detection result, indicating that a sound source is currently passing by the microphone array 102.

Appropriate values for the number of time windows N , and the reference value n_t will depend upon the approximate motion speed which can be expected for the sound sources, and upon the distance L of the path of the sound sources from the microphone array 102. If for example L is 10 meters and the speed of motion of the sound sources can be expected to be approximately 40 km/hour, then a value for the number of time windows N in the range 10 to 40 and a value for the reference value n_t of approximately 5 to 10 would be suitable

Fig. 7 is a flow diagram of an example of an algorithm which could be used to implement the

operation of this embodiment. To simplify the diagram, it is assumed that an estimated direction is obtained in every time window, although in practice this may not always occur. As shown, incrementing of count values m and n that are to be compared with the time window number N and the threshold value n_t , respectively, begins when it is judged in step S3 that an estimated sound source direction has come within the passage detection range P_n specified by the direction range setting section 211, i.e., when the sound source has reached or exceeded the direction θ_r shown in Fig. 6. When this is found, the loop of steps S5 to S11 is repetitively executed until it is found that the threshold number of in-range estimated directions n_t has been reached before the number of successive time windows N has been reached (in which case a passage detection result is generated) or the number of time windows N is attained without that threshold number n_t having been reached.

The configuration of the second embodiment described above is intended for application to sound sources which are travelling along a single direction, i.e., that indicated by the arrow on the path traveled by the sound source 101 in Fig. 6. In

the case of passage detection of sound sources travelling on adjacent parallel paths but in opposite directions (e.g., as in the case of traffic along a two-way street) a single microphone array 102 would be utilized, however a second passage detection section 216 would be provided for detecting the passing of sound sources which travel in the second motion direction, as described in detail hereinafter with reference to an eighth embodiment.

Third Embodiment

A third embodiment will be described referring to the general system block diagram of Fig. 8. As shown, this is formed of a microphone array 102, sound source direction estimation section 116 and passage detection section 216, together with a data buffer 307, a data extraction section 308 and a recording apparatus 309. The sound source direction estimation section 116 can have the first configuration shown in Figs. 2A, 2B, whereby each output estimated direction is derived based upon frequency-based averaging and upon time-based averaging applied over a plurality of successive time windows, or can have the alternative configuration shown in Fig. 4, with only frequency-

based averaging being applied. The passage detection section 216 has the configuration and operation described above for the preceding embodiment.

5 The operation of this embodiment is as follows. Data expressing respective passage detection results are supplied from the passage detection section 216 to the data extraction section 308, and the audio signal produced from one of the microphones of the
10 array 102, which will be assumed to be the microphone in the No. 1 array position, is supplied to the data buffer 307, i.e., as successive digital data samples. (For simplicity of description, analog-to-digital converters for converting the
15 microphone output signals to digital signals are omitted from the drawings). It would be equally possible to utilize a combination of output signals from more than one of the microphones. When a passage detection result is supplied to the data
20 extraction section 308, it responds by reading out the data currently held in the data buffer 307 and supplying the data to the recording apparatus 309 to be recorded thereby.

 The capacity of the buffer 307 is such that at
25 any one time, the buffer holds an amount of audio

data that has been generated within a specific time interval T_b . When the data extraction section 308 receives a passage detection result indicating that a sound source is passing by the microphone array 102, the data extraction section 308 reads out the contents of the data buffer 307 and transfers these to the recording apparatus 309, to be recorded. In that way, a signal expressing the sound emitted from a sound source which is currently passing the microphone array 102, covering a time interval of duration T_b and extending up to the time point of obtaining the passage detection result, is supplied to the recording apparatus 309. The sound produced from a moving sound source during a specific time interval, as it moves past the microphone array 102, can thereby be obtained and recorded.

Thus with this embodiment, in the same way as for the second embodiment, the direction of a sound source can be estimated and the passing of that sound source can be detected, while in addition, the output signal from at least one of microphone is utilized to monitor and record the sound produced from the passing sound source, with these operations being triggered by generation of a passage detection result.

The processing sequence of the above operation is shown in the flow diagram of Fig. 9.

Fourth Embodiment

A fourth embodiment will be described referring first to the general system block diagram of Fig. 10. As shown, this is formed of a microphone array 102 and a sound source direction estimation section 116, together with a stationary sound source derivation section 408 which is formed of a moving average calculation section 405, a variance calculation section 406 and a stationary sound source detection section 407. The sound source direction estimation section 116 can have the first configuration shown in Figs. 2A, 2B, whereby each output estimated direction is derived based upon frequency-based averaging and upon time-based averaging applied over a plurality of successive time windows, or can have the alternative configuration shown in Fig. 4, with only frequency-based averaging being applied.

The data expressing respective estimated directions produced from the sound source direction estimation section 116 are inputted to each of the moving average calculation section 405 and the variance calculation section 406 of the stationary

sound source derivation section 408. The
respective outputs from the moving average
calculation section 405 and variance calculation
section 406 are inputted to the stationary sound
5 source detector 407. When occurrence of a
stationary sound source is detected, the stationary
sound source detector 407 generates output data
indicative of that detection result.

In the stationary sound source detection
10 section 408, the moving average calculation section
405 calculates the moving average of successive sets
of sound source estimated directions produced from
the sound source direction estimation section 116,
in each of successive fixed-duration observation
15 intervals, and the variance calculation section 406
calculates the variance of the successive estimated
directions. The stationary sound source detector
407 judges that a sound source is stationary if the
variance that is calculated by the variance
20 calculation section 406 is found to be lower than a
reference value. If a moving average of the
estimated directions is found to be outside a
predetermined range of possible directions for the
sound source, then the corresponding variance value
25 is discarded.

In that way, the stationary sound source detector 407 can reliably detect the occurrence of a stationary sound source that is located within the range of directionality of the microphone array 102.

5 In addition, the estimated direction which is produced from the sound source direction estimation section 116 at the time when such detection occurs represents the direction of that stationary sound source.

10 Fifth Embodiment

A fifth embodiment will be described referring to the general system block diagram of Fig. 11 and the flow diagram of Fig. 12. The embodiment is formed of a microphone array 102, a sound source direction estimation section 116, a passage
15 detection section 216, each of which operates as described hereinabove for correspondingly numbered sections of preceding embodiments, and a motion direction derivation section 509. The motion
20 direction derivation section 509 is formed of a estimated direction buffer 505, a prior-to-passage direction derivation section 506, a subsequent-to-passage direction derivation section 507 and a motion direction detection section 508. In the same
25 way as described for the second embodiment,

estimated directions obtained by the sound source
direction estimation section 116 in successive time
windows are supplied to the passage detection
section 216, with a passage detection result being
5 generated by the passage detection section 216 each
time it is detected that a sound source has entered
the passage detection range Pn. In the motion
direction derivation section 509, the estimated
direction buffer 505 is configured to have a storage
10 capacity corresponding to an appropriate number of
successively obtained estimated directions.

When data expressing a passage detection
result (indicated as Pd in Fig. 11) is generated by
the passage detection section 216, this is supplied
15 to the prior-to-passage direction derivation section
506 and the subsequent-to-passage direction
derivation section 507. The subsequent-to-passage
direction derivation section 507 responds by
transferring to the motion direction detection
20 section 508 the estimated direction which was most
recently set into the buffer 505, or the most
recently obtained estimated direction produced from
the passage detection section 216. The prior-to-
passage direction derivation section 506 responds to
25 the passage detection result by transferring to the

motion direction detection section 508 an estimated direction which was obtained prior to that which is transferred by the subsequent-to-passage direction derivation section 507, such as the oldest estimated
5 direction in the contents of the buffer 505.

Designating these respective estimated directions extracted by the prior-to-passage direction derivation section 506 and subsequent-to-passage direction derivation section 507 as θ_{bf} and θ_{af} , the
10 motion direction detection section 508 then obtains the sign of the difference between θ_{bf} and θ_{af} , with that sign indicating the direction of motion of the sound source. For example in the case of Fig. 11,
15 if $(\theta_{af} - \theta_{bf})$ is positive, then this indicates that the motion direction of the sound source is along the direction of the arrow.

Fig. 12 is a flow diagram showing the processing executed with the above form of operation of this embodiment.
20

In the above description of the fifth embodiment, a data buffer 505 such as a FIFO (first-in first-out) buffer is used to obtain an estimated direction that was produced at a time prior to the point in time at which a passage detection result
25 was generated. However it will be understood that

as an alternative form of implementing this
embodiment, it would be equally possible to register
the estimated direction that is obtained at the time
point when a passage detection result is generated,
5 to wait for a predetermined number of time window
periods and obtain the estimated direction which is
produced at that time point from the passage
detection section 216, and subtract the two
estimated directions thus obtained from one another,
10 to thereby obtain the motion direction of a sound
source as indicated by the sign of the subtraction
result.

As a further alternative form of implementing
this embodiment, it would be possible to use a
15 buffer as described for the estimated direction
buffer 505 to obtain an estimated direction which
was produced by the passage detection section 216 at
a time point prior to the time point when a passage
detection result is generated, to wait for a
20 predetermined number of time window periods after a
passage detection result is generated and then
obtain the estimated direction which is produced at
that time point from the passage detection section
216, and subtract the earliest and most recent
25 estimated directions thus obtained from one another,

to thereby obtain the motion direction of a sound source as indicated by the sign of the subtraction result.

It can thus be understood that whichever of the above alternatives is employed, each is based upon using a passage detection result as an indication that a sound source has traversed a specific range of directions, and so is currently moving past the microphone array 102.

Sixth Embodiment

A sixth embodiment will be described referring to the general system block diagram of Fig. 13 and the flow diagram of Fig. 14. The embodiment is formed of a microphone array 102, a sound source direction estimation section 116, a passage detection section 216, each of which operates as described hereinabove for correspondingly numbered sections of preceding embodiments, and a velocity derivation section 609. The velocity derivation section 609 is formed of a data buffer 605, a motion interval calculation section 606, an angular amount determining section 607 and a velocity derivation section 609. The velocity derivation section 609 operates on data provided from the sound source direction estimation section 116 and passage

detection section 216 to determine the velocity of a sound source which moves past the microphone array 102.

The operation is as follows. The successive
5 estimated directions of a sound source, produced from the sound source direction estimation section 116, are supplied to the passage detection section 216 and to the data buffer 605 of the velocity derivation section 609. The data buffer 605 is
10 capable of storing data expressing a fixed plurality of successively obtained estimated directions of a sound source, with the number of estimated directions being predetermined in accordance with an anticipated typical or maximum velocity of the sound
15 sources and in accordance with the magnitude of a fixedly predetermined angular amount Ω (described hereinafter). Successively obtained estimated directions of a sound source, corresponding to respective time windows, are sequentially written
20 into the buffer 605. When the passage detection section passage detection section 216 detects that a sound source is moving past the microphone array 102 (as indicated by generation of a passage detection result), then the contents of the data buffer 605 at

that time are read out by the motion interval calculation section 606.

The motion interval calculation section 606 then determines, based on the set of past estimated
 5 directions respectively corresponding to successive time points which are thereby obtained, the length of time T (i.e., expressed as a number of time window periods) which was required for the sound source to move through a preset angular amount
 10 (i.e., range of successive directions with respect to the microphone array 102) designated as Ω , which is specified by the angular amount determining section 607.

The velocity detection section 608 then
 15 calculates the angular velocity at which the sound source is moving, based on the angular amount Ω and the value obtained for T, by using equation (8) below:

$$\omega = \Omega/T \quad \dots\dots\dots(8)$$

20 The distance L between the microphone array 602 and the sound source 601 is known beforehand. The velocity detection section 608 calculates the linear velocity V of the sound source by using equation (9) below:

$$25 \quad V = L \tan(\omega) \quad \dots\dots\dots(9)$$

Data expressing the calculated velocity value are the outputted from the velocity detection section 608.

Fig. 14 illustrates the processing sequence executed with this embodiment, in the case of the form of operation described above.

Although the sixth embodiment has been described above for the case in which the angular velocity of a sound source is calculated by measuring the amount of time required for the sound source directions to move through a predetermined angular amount, it would be equally possible to predetermine a fixed time interval, and to use the contents of the buffer 605 to obtain the angular amount by which the sound source directions have changed within that predetermined time interval.

Furthermore, although the sixth embodiment has been described above for the case in which the angular velocity of a sound source is calculated on the basis of an amount of change in the sound source directions which occurred prior to the time point at which a passage detection result is generated, it would be equally possible to calculate the angular velocity of a sound source on the basis of an amount of change in the sound source directions which

occurs subsequent to the time point at which a
passage detection result is generated. That is to
say, it would be possible for example to measure the
amount of angular change in the successively
5 obtained estimated directions which occurs during a
fixed time interval following the time point at
which a passage detection result is generated.

Thus with this embodiment, not only can the
passage of a sound source be detected, but also the
10 velocity at which the sound source is passing can
also be derived, by using the passage detection
result as a trigger for initiating velocity
derivation processing.

Seventh Embodiment

15 A seventh embodiment will be described
referring to the general system block diagram of
Fig. 15 and the flow diagram of Fig. 16. The
embodiment is formed of a microphone array 102, a
sound source direction estimation section 116, a
20 passage detection section 216, each of which
operates as described hereinabove for
correspondingly numbered sections of preceding
embodiments, and a directivity control section 706.
The directivity control section 706 is formed of a
25 directivity controller 704, an adder 705, and an

array of delay elements 703. The array of delay elements 703 respectively correspond to, and receive respective audio signals from, the microphones of the microphone array 102.

5 The directivity control section 706 operates on the output signals produced from the microphone array 102 and data provided from the sound source direction estimation section 116 and passage detection section 216 to obtain a monitoring signal
10 which expresses the sound that is being emitted from a sound source as the sound source moves past the microphone array 102. Since the sound emitted from each of the sound sources will be monitored at a time when the sound sources are at substantially the
15 same distance from the microphone array, such a monitoring signal can be used for example to estimate the respective levels of sound power which are produced by successive sound sources.

 In the directivity control section 706, the
20 direction controller 704 sets the respective delays of the delay elements 703 in response to a passage detection result being produced from passage detection section 216, in the following manner.
 When a sound source is detected as passing by the
25 microphone array 102 (as indicated by generation of

a passage detection result by the passage detection section 216) then designating the estimated direction obtained for the sound source at the time point when a passage detection result is generated as θ_s , the direction controller 704 calculates the vector $d[m]$ ($m=1, 2, \dots, M$) by using equation (10) below:

$$d[m] = [1, e^{-j\omega\tau}, e^{-j\omega^2\tau}, \dots, e^{-j\omega(M-1)\tau}]^T \quad \dots(10)$$

here, τ is defined by equation (11) below, in which c denotes the speed of sound:

$$\tau = (d \sin \theta_s) / c \quad \dots(11)$$

The elements of the vector $d[m]$ are set as the respective delay values of the M delay elements 703. By setting the delay values in that way, the respective delayed audio signals produced from the delay elements are synchronized in phase with the sound source signal which arrives along the direction θ_s , i.e., by summing the outputs from the delay elements in the adder 705, a monitoring signal is obtained with directivity applied along the θ_s direction. Thus, the level of sound being emitted from the sound source which is currently at the direction θ_s can be selectively obtained, i.e., the

sound produced from a specific moving sound source can be monitored.

Hence, the directivity control section 706 produces a monitoring signal that is obtained as a combination of the output signals from the microphone array 102, with an appropriate directivity applied.

Thus with the seventh embodiment, in addition to obtaining successive estimated directions of a sound source in relation to the microphone array 102, and judging when the sound source is currently passing by the microphone array 102, the level of sound emitted from that sound source can be monitored by appropriately orienting the effective directivity of the microphone array 102 in the direction of that sound source, with detection of passing of the sound source being used as a trigger for initiating this orienting of the directivity. With this embodiment therefore, using only a single array of microphones, it becomes possible to selectively monitor a sound emitted from a specific sound source which is at an arbitrary direction from the microphone array.

Various ways of utilizing the monitoring signal produced by this embodiment could be envisaged, for

example setting successive pluralities of monitoring signal samples in a data buffer (not shown in the drawings), and reading out the current buffer contents after a fixed time interval has elapsed following a time point at which a passage detection result is generated. In that way, the condition of the monitoring signal while each specific sound source is passing the microphone array 102 can be obtained.

10 The processing sequence of this embodiment is illustrated in the flow diagram of Fig. 16, assuming the case in which such an output buffer is utilized.

Eighth Embodiment

15 With the preceding embodiment, it is necessary for the directivity controller 704 to establish a new set of delay values for the set of delay elements 703, each time that a passage detection result is obtained for a sound source from the passage detection section 216 together with an estimated direction produced from the sound source direction estimation section 116, in order to align the effective directivity of the microphone array 102 with the current estimated direction of the sound source. These estimated directions may vary substantially, depending upon the respective

20

25

velocities of the sound sources, etc. For example, if a sound source is moving very slowly, then the aforementioned threshold number of in-range estimated directions n_t may be reached at a point when the sound source has moved through only an initial small part of the passage detection range P_n . If on the other hand a sound source is moving very rapidly, so that its direction with respect to the microphone array changes by a large angular amount in each time window, then the sound source may have almost reached the final direction θ_f of the passage detection range P_n at the point when the threshold number of in-range estimated directions n_t has been reached for that sound source.

In the case of sound sources which are moving with high velocity and are separated from one another by relatively small distances, e.g., vehicles travelling on an expressway, it may be impossible (due to limitations of system performance) to synchronize the operation of establishing an updated set of delay values for the delay elements 703 with the time point at which a new passage detection result is generated. An eighth embodiment will be described in the following, which is designed to overcome this

problem. Fig. 17 shows the general configuration of this embodiment. This has basic similarities to the preceding embodiment, being formed of a sound source direction estimation section 116, a passage
5 detection section 216, two data extractor sections 814, 817 and corresponding data buffers 813, 816, which receive data values produced from a directivity control section 706A and a directivity control section 706B respectively. With this
10 embodiment, the output signals from the microphone array 102 are supplied not only to the sound source direction estimation section 116 and the directivity control section 706A but also to the directivity control section 706B.

15 Each of the directivity control sections 706A, 706B performs a similar function to that of the directivity control section 706 of the preceding embodiment shown in Fig. 15. However in each of the directivity control sections 706A, 706B, the
20 respective delay values assigned to the set of delay elements are fixedly predetermined, such as to provide directivity along a direction θ_1 in the case of the directivity control section 706A and along a direction θ_2 in the case of the directivity control
25 section 706B. Taking the directivity control

section 706A for example, each time a new digital data sample expressing the sound level received along the direction θ_1 is generated by the directivity control section 706A (i.e., as part of a monitoring signal produced by the directivity control section 706A as a combination of the output signals from the delay elements 703 with a specific directivity applied) it is written into the buffer 813 under the control of the data extraction section 814.

The directivity control section 706B, buffer 816 and data extraction section 817 operate in the same manner, with respect to the direction θ_2 .

The basic principles of the embodiment can be understood by referring to the conceptual diagram of Fig. 19. As shown, a first predetermined range of directions R1 is defined in relation to the microphone array 102, which is centered on the direction θ_1 , while a second range of directions R2 is centered on the direction θ_2 . The aforementioned predetermined range of directions Pn which is used by the passage detection section 216 to judge whether passage detection has occurred (i.e., as described hereinabove, whereby a count is performed

of the number of times that estimated directions within that range are obtained for a sound source, and passage detection is judged to have occurred if the number counted within N successive time window periods is above the threshold number n_t) extends from an initial direction θ_r to a final direction θ_f . In Fig. 19, the threshold number of in-range estimated directions n_t might be reached for the sound source 100 when it has reached the position shown, within the range R1. The data extraction section 814 accordingly reads out from the data buffer 813 the sound level data currently held in that buffer, i.e., data expressing variations in received sound level as obtained from directivity along the direction θ_1 . Similarly, The threshold number n_t might be reached for the sound source 101 when it has reached the position shown, within the range R2. The data extraction section 817 accordingly reads out from the data buffer 816 the data currently held in that buffer, i.e., data expressing variations in received sound level as obtained from directivity along the direction θ_2 .

Read-out from the buffers 814, 817 are performed under the control of the data extraction

sections 814, 817 at appropriate timings, e.g., in accordance with the requirements of an external apparatus which receives the sound level data.

It can thus be understood that with this
5 embodiment, the disadvantage of the preceding
embodiment is overcome, since the delay values of
the sets of delay elements in the directivity
control sections are fixedly predetermined, so that
updating of the delay values of the delay elements
10 703 is not performed. However for each sound
source, a monitoring data expressing variations in
received sound level corresponding to that sound
source will be obtained based on a microphone array
directivity which is at least close to the direction
15 of the sound source at the time of monitoring the
received sound level. Hence monitored sound level
values which are approximately correct can be
obtained even for the case in which some of the
sound sources may be moving at high velocity.

20 It will be further understood that although
with this embodiment only two directivity control
sections are utilized, it would be possible to
achieve higher accuracy of sound level measurement,
or increase the range of sound source velocities for
25 which measurement can be performed, by increasing

the number of directivity control sections, to thereby increase the number of ranges of angular directions such as R1 and R2 in Fig. 19.

Fig. 18 is a flow diagram illustrating the
5 basic principles of operation of this embodiment.

With the eighth embodiment, if for example two
of the sound sources are travelling at a
substantially identical high velocity and are
separated by only a small distance, then the
10 respective passage detection results for each of
these sound sources would be successively produced
from the passage detection section 216 within a very
short time interval. Thus, the minimum amount of
time which must elapse before data are read out from
15 a buffer 813 or 816 after being written therein (in
response to a passage detection result) is
determined by the maximum velocity which will be
attained by sound sources and by the minimum
distance between successive sound sources.
20 Designating the separation between the two sound
sources (expressed as a distance along the line of
travel) as S and their common velocity as V , it is
necessary to perform data read-out from such a
buffer (after data write-in as a result of detection
25 of passage of the first of these sound sources)

within a time interval that is shorter than approximately S/V , in order to ensure that the sound level data for the first sound source will not be lost due to write-in of sound level data for the second sound source.

Fig. 20 is a conceptual diagram for illustrating how two passage detection sections 216A, 216B, each configured in a similar manner to the passage detection section 216, could be used to derive passage detection data for sound sources travelling in opposing directions, such as motor vehicle traffic on a two-way street, and supply respective passage detection results to two sets of directivity control sections 807A, 811A and 807B, 811B. In Fig. 22, the microphone array 102 is disposed adjacent to such a two-way street, in which traffic flows along a first path 805, which will be referred to as the upstream path, and a second path 804 which will be referred to as the downstream path. The microphone output signals from the microphone array 102 are supplied to the sound source direction estimation section 116 and to the directivity control sections 807A, 811A as in the embodiment of Fig. 17, and also to the second pair of directivity control sections 807B, 811B. The

5 The passage detection results from the passage
detection section 216A (which detects only motion
through a passage detection range along the upstream
path 805) and each estimated direction corresponding
to such a passage detection result, are supplied to
10 the directivity control sections 807A, 811A, which
function in a similar manner to the directivity
control sections 807, 811 in Fig. 17. The passage
detection results from the passage detection section
216B (which detects only motion through the passage
15 detection range from θ_r to θ_f shown in Fig. 20 along
the downstream path 804) and each estimated
direction corresponding to such a passage detection
result, are supplied to the directivity control
sections 807B, 811B. The range of directions used by
20 the directivity control sections 807A, 811A for
passage detection in the upstream path 805 (omitted
from Fig. 22) can be as shown in Fig. 19, i.e.,
extending from θ_r to θ_f in that diagram.

In that way, a single microphone array 102 and single sound source direction estimation section 116

can be used in implementing sound source passage detection and microphone array directivity control for both of the motion paths 804, 805.

Fig. 21 is a partial flow diagram (in which
5 omitted steps S2, S3 are identical to those of the
flow diagram of Fig. 7) for illustrating the
processing sequence executed by each of the passage
detection sections 216A, 216B. It will be understood
that the condition step S8, in this case, must
10 actually determine whether the current estimated
direction (or one of these directions, if a
plurality are generated for one time window by the
passage detection section 216):

(a) is greater than (i.e., further advanced
15 towards the final direction θ_f) or equal to the
estimated direction obtained for the preceding time
window, and

(b) is within the passage detection range P_n ,
and

20 (c) differs from the estimated direction
obtained for the preceding time window by no more
than a predetermined maximum amount.

Conditions (a) and (c) above serve to ensure
that estimated directions obtained for any sound
25 source travelling in the opposite direction to the

desired sound source will be ignored. The maximum amount which is specified in that condition (c) will depend upon the minimum amount of distance anticipated between successive sound sources, and
5 the range of velocities anticipated for the sound sources

Although the present invention has been described in the above referring to specific embodiments, it is to be understood that various
10 modifications to these embodiments or alternative forms of implementing the embodiments could be envisaged, which fall within the scope claimed for the invention in the appended claims.

15

20

25